AMENDMENTS TO THE CLAIMS

This listing of claims replaces all prior versions, and listings, of claims in the application:

Listing of Claims

1-34. (Cancelled)

35. (New) User equipment in a communications system, said user equipment comprising:

means for receiving data packets comprising bursty audio data over said communications system;

a size variable playout buffer configured for temporarily storing said data packets; and.

means for adapting the playout buffer size based on the audio burst length.

36. (New) The user equipment according to claim 35, further comprising:

means for analyzing information associated with said packets; and

means for determining said audio burst length based on said analyzed information.

37. (New) The user equipment according to claim 36, wherein said analyzing means comprises means for determining a number of bits in said data packets from an audio burst start identifier to an audio burst stop identifier, and said length determining means is configured for determining said audio burst length based on said determined number of bits.

38. (New) The user equipment according to claim 36, wherein said analyzing means comprises means for calculating a number of data packets received by said receiving means from a first data packet comprising an audio burst start identifier to a second data packet comprising an audio burst stop identifier, and said length determining means is configured for determining said audio burst length based on said calculated number of data packets.

39. (New) The user equipment according to claim 36, wherein said analyzing means comprises means for determining a total releasing time comprising a time of releasing a data packet comprising an audio burst start identifier from said playout buffer to a time of releasing a data packet comprising an audio burst stop identifier from said playout buffer, and said length determining means is configured for determining said audio burst length based on said determined total releasing time.

40. (New) The user equipment according to claim 35, further comprising means for determining an average length of multiple audio bursts, wherein said size adapting means is configured for determining said playout buffer size based on said determined average length.

41. (New) The user equipment according to claim 40, wherein said average length determining means is configured for determining a weighted average length of said multiple audio bursts.

42. (New) The user equipment according to claim 40, further comprising means for determining a number of audio bursts that are to be used by said average length determining means for said average length determination based on the audio burst length.

43. (New) The user equipment according to claim 35, further comprising means for estimating a variation of transmission delay for said data packets from a transmitting node, wherein said size adapting means is configured for adapting said playout buffer size based on said estimated delay variation.

44. (New) The user equipment according to claim 35, wherein said size adapting means is configured for setting said playout buffer size at a first size if said audio burst length is according to a first length value and setting said playout buffer size at a second relatively larger size if said audio burst length is larger than said first length value.

45. (New) The user equipment according to claim 35, further comprising a client configured for supporting Push to talk over Cellular (PoC) services in a packed based radio communications system, and said size adapting means and said playout buffer are configured in said PoC client.

46. (New) A buffer controller for an associated playout buffer that is configured for temporarily storing data packets comprising bursty audio data, said controller comprising:

means for analyzing information associated with said data packets for determining the audio burst length; and

means for adapting said size of said playout buffer based on said determined audio burst length.

47. (New) The controller according to claim 46, wherein said analyzing means comprises means for determining a number of bits in said data packets from an audio burst start identifier to an audio burst stop identifier, and said controller comprises means for determining said audio burst length based on said determined number of bits.

48. (New) The controller according to claim 46, wherein said analyzing means comprises means for calculating a number of data packets stored in said playout buffer from a first data packet comprising an audio burst start identifier to a second data packet comprising an audio burst stop identifier, and said controller comprises means for determining said audio burst length based on said calculated number of data packets.

49. (New) The controller according to claim 46, wherein said analyzing means comprises means for determining a total releasing time that comprises a time of releasing a data packet comprising an audio burst start identifier from said playout buffer to a time of releasing a data packet comprising an audio burst stop identifier from said playout buffer, and said controller comprises means for determining said audio burst length based on said determined total releasing time.

50. (New) The controller according to claim 46, further comprising means for determining an average length of multiple audio bursts, wherein said size adapting means is configured for determining said playout buffer size based on said determined average length.

51. (New) The controller according to claim 50, wherein said average length determining means is configured for determining a weighted average length of said multiple audio bursts.

52. (New) The controller according to claim 50, further comprising means for determining a number of audio bursts that are to be used by said average length determining means for said average length determination based on the audio burst length.

53. (New) The controller according to claim 46, further comprising means for estimating a variation of transmission delay for said data packets from a transmitting node over a communications system, wherein said size adapting means is configured for adapting said playout buffer size based on said estimated delay variation.

54. (New) The controller according to claim 46, wherein said size adapting means is configured for setting said playout buffer size at a first size if said audio burst length is according to a first length value and setting said playout buffer size at a second relatively larger size if said audio burst length is larger than said first length value.

55. (New) The controller according to claim 46, wherein said buffer controller and said associated playout buffer are provided in a client configured for supporting Push to talk over Cellular (PoC) services in user equipment in a packet based radio communications system.

56. (New) A method of controlling a playout buffer that temporary stores data packets comprising bursty audio data received over a communications system, said method comprising the steps of :

determining the audio burst length; and

adapting said size of said playout buffer based on said determined audio burst length.

57. (New) The method according to claim 56, wherein said length determining step comprises the step of determining a number of bits in said data packets from an audio burst start identifier to an audio burst stop identifier.

58. (New) The method according to claim 56, wherein said length determining step comprises the step of calculating a number of received data packets from a first data packet comprising an audio burst start identifier to a second data packet comprising an audio burst stop identifier.

59. (New) The method according to claim 56, wherein said length determining step comprises the step of determining a total releasing time comprising a time of releasing a data packet comprising an audio burst start identifier from said playout buffer to a time of releasing a data packet comprising an audio burst stop identifier from said playout buffer.

60. (New) The method according to claim 56, further comprising the step of determining an average length of multiple audio bursts, wherein said size adapting step comprises the step of adapting said playout buffer size based on said determined average length.

61. (New) The method according to claim 60, wherein said average length determining step comprises the step of determining a weighted average length of said multiple audio bursts.

62. (New) The method according to claim 61, wherein a weight for a first audio burst is larger than a weight for a second audio burst, where the audio data of said second audio burst being received at said playout buffer at an earlier instance than the audio data of said first audio burst.

63. (New) The method according to claim 60, further comprising the step of determining a number of audio bursts that are to be included in said average length determination based on the audio burst length.

64. (New) The method according to claim 56, further comprising the step of estimating a variation of transmission delay for said data packets from a transmitting node, wherein said adapting step comprises the step of adapting said playout buffer size based on said estimated delay variation.

65. (New) The method according to claim 56, wherein said playout buffer size is set at a first size if said determined audio burst length is according to a first length value and said playout buffer size is set at a second relatively larger size if said determined audio burst length is larger than said first length value.

66. (New) A communications system, comprising:

a transmitting node transmitting data packets comprising bursty audio data; and

a receiving node adapted for receiving said transmitted data packets and comprising:

a size variable playout buffer configured for temporarily storing said packets; and

means for adapting the playout buffer size based on the audio burst length.

67. (New) The system according to claim 66, wherein said transmitting node comprises means for introducing, into data packets, an audio burst start identifier and an audio burst stop identifier, and said receiving node comprises means for

determining said audio burst length based on said audio burst start identifier and said audio burst stop identifier.

68. (New) The system according to claim 66, wherein said communications system is a packet based radio communications system supporting Push to talk over Cellular (PoC) services, said transmitting node is first user equipment comprising a first PoC client and said receiving node is second user equipment comprising a second PoC client, said size adapting means and said playout buffer being provided in said second PoC client.

* * *